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(71) Applicant:
Siemens Business Communication Systems, Inc.
Santa Clara, CA 95052 (US)

(72) Inventors:
- Shaffer, Shmuel
Palo Alto, CA 94301 (US)
- Beyda, William J.
Cupertino, CA 95014 (US)

(74) Representative: Allen, Derek
Siemens Group Services Limited,
Intellectual Property Department,
Siemens House,
Oldbury
Bracknell, Berkshire RG12 8FZ (GB)

(54) **Alternative telephone call routing system**

(57) A telecommunication system includes a source private branch exchange (12) that transmits telephone calls from a source to a destination private branch exchange (26) over a public switched telephone network (18). As an alternative to transmitting the calls over the public switched telephone network, the private branch exchange is coupled to a telephony Internet server 30 that can transmit telephone calls over a global wide area computer network such as the Internet (32). The private branch exchange (12) queries the destination private branch exchange (26) to determine if it similarly coupled to a telephony Internet server. If so, the bandwidth used by the telephony Internet server will allow the transmission of the telephone call, the call is routed over the Internet (32) to the intended recipient.

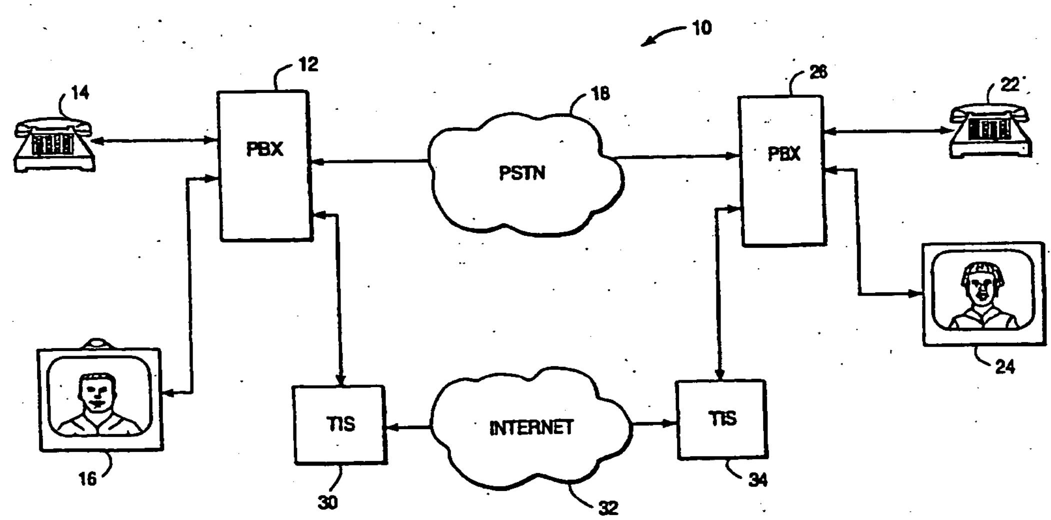

FIG_1

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Description

Field of the Invention

[0001] The present invention relates to communication systems, and in particular to systems for selecting alternative routes for telephone calls.

Background of the Invention

[0002] As business is conducted over ever expanding geographical areas, the use of telecommunication services to transmit voice and data signals is continually increasing. As a result of the increased use of these services, the cost for such services becomes a significant part of the cost of doing business. Therefore, most businesses are continually looking for ways to reduce their communication costs.

[0003] One known method of obtaining "free" telephone calls is to transmit the calls via a global wide area computer network such as the Internet. In effect, these calls are not free because the user leases the telephone lines that are used to provide their Internet access, however, calls placed over the Internet are not separately billed.

[0004] In the past, it has not been possible to seamlessly integrate the Internet as an alternative mute for transmitting telephone calls because there is no way of knowing whether an intended recipient has the ability to receive calls.

[0005] Given the shortcomings in the prior art, there is a need for a telephone communication system that can reduce communication costs by automatically determining when it is possible to transmit calls on the Internet to utilize excess bandwidth.

Summary of the Invention

[0006] To reduce the cost associated with transmitting telephone calls over a public switched telephone network (PSTN), the present invention is a communication system that transmits calls from a source private branch exchange or central office to a destination private branch exchange or central office. The source private branch exchange is coupled to a telephony Internet server that can transmit a call over a global wide area computer network such as the Internet. To determine whether the call can be transmitted over the Internet, the source private branch exchange transmits a message to the destination private branch exchange over the PSTN to determine whether it is similarly equipped with a telephony Internet server. If so, and the bandwidth available on the Internet will accommodate an additional call, then the telephone call is routed to the Internet.

[0007] The quality of the call placed on the Internet is continually monitored. If the quality drops below a predetermined threshold, the call is rerouted from the Internet back to the public switched telephone network.

Brief Description of the Drawings

[0008] The foregoing aspects and many of the attendant advantages of this invention will become more readily appreciated as the same becomes better understood by reference to the following detailed description, when taken in conjunction with the accompanying drawings, wherein:

FIGURE 1 is a block diagram of a communication system in accordance with the present invention; and

FIGURE 2 is a flow chart of the steps performed by the present invention to select the most appropriate path to route a telephone call.

Detailed Description of the Preferred Embodiment

[0009] The present invention is a communication system that can automatically determine whether to route a telephone call on a public switched telephone network or on a alternative path, such as the Internet, in order to reduce communication charges.

[0010] As shown in FIGURE 1, the communication system 10, according to the present invention, includes a source private branch change (PBX) 12 that connects a telephone call between a number of user input devices and a public switched telephone network (PSTN) 18. The input devices may be standard telephones 14, a video conferencing system 16 or other types of communication systems such a facsimile machine, etc. Calls from a user input device are typically routed by the PBX 12 on the public switched telephone network 18 to a intended receiver. The intended receiver may be a conventional telephone 22 or a corresponding video conferencing system 24, facsimile machine, etc. The intended receivers are generally coupled to the public switched telephone network 18 through a destination private branch exchange PBX 26.

[0011] As described above, each time a user places a call on the public switched telephone network 18, they are charged for the use of the service. An alternative method of transmitting a telephone call is through the use of a global wide area computer network such as the Internet. To transmit these calls, telephony Internet server 30 is coupled to the PBX 12. The telephony Internet server receives a digitized telephone signal, compresses the signal, and arranges the compressed signal into a series of data packets. An Internet address is added to each packet and the packets are transmitted over the Internet 32 to a receiving telephony Internet server 34, that is coupled to the receiving PBX 26. At the receiving telephony Internet server, the packets are decompressed, combined back into a serial data stream, and supplied to the PBX 26.

[0012] To reduce the cost of communication services,

the communication system of the present invention determines when it is possible to mute a telephone call on the Internet 32 rather than on the PSTN 18. In particular, if a desired recipient's PBX 26 is equipped with a telephony Internet server, and the bandwidth being used by such a server can handle the additional traffic, then a telephone call can be muted on the Internet to avoid paying the additional charges that would be incurred if the call were transmitted on the PSTN 18.

[0013] FIGURE 2 is a flow chart of the steps 50 performed by the PBX 12 shown in FIGURE 1 in order to determine whether a call should be muted on the PSTN 18 or on the Internet 32. Beginning with a step 52, the source PBX receives a telephone number of an intended recipient from a user input device such as a telephone, video conferencing system, facsimile machine, etc. At a step 54, the source PBX determines whether the telephone number received is in the list of recently placed Internet calls. If the answer to step 54 is no, then the PBX 12 begins to place the call on the PSTN 18 using an ISDN or other similar digital format.

[0014] As the call is being set up, the source PBX transmits an information element field that indicates the source PBX has a telephony Internet server with the ability to route the call over the Internet. This information element field is received by the destination PBX and decoded. The destination PBX then responds if it is similarly equipped with a telephony Internet server and if so includes with the response the Internet address of its telephony Internet server. At step 60, the sending PBX determines whether a reply has been received with an address of a destination telephony Internet server within a predetermined time limit. This time limit may be fixed or variable depending on who is attempting to place a call. For example, if the call is placed from an executive phone, then the time limit used before connecting a call on the PSTN may be shorter than the time limit used for calls that originate from the mailroom. If the answer to step 60 is no, then the call set up is completed on the PSTN at a step 62 in a conventional manner.

[0015] If the answer to either the step 54 or 60 is yes, then the PBX forwards the address of the destination telephony Internet server to the telephony Internet server that is coupled to the source PBX at a step 62. The telephony Internet server uses this address to send the packetized telephone call to the intended receiver on the Internet at a step 64. Calls may be immediately rerouted from a PSTN to the Internet upon the determination on that the receiver is equipped with a telephony Internet server. If a response is received indicating that the call can be routed on the Internet after the call has already been set up on the PSTN, then the call can be switched to the Internet after a time period equal to the minimum billing increment on the PSTN. For example, if a call is initially set up on the PSTN and the PSTN bills in one minute increments, the call would be switched to the Internet at the end of the minute

[0016] At any time the quality of the data transmission carried by the telephony Internet servers may degrade such that the call cannot be properly transmitted on the Internet. Therefore, at a step 66, the source PBX queries the telephony Internet server regarding the quality of the call placed on the Internet. Typically, quality is measured by the number of data packets that are transmitted in a given amount of time and the delay introduced by the telephony Internet servers and the network that extends between them to send the packets. Methods for establishing a level of quality on a packetized data network such as the Internet are considered well known to those of ordinary skill in the art and therefore need not be discussed in further detail.

[0017] At a step 68, the sending PBX determines whether the quality of the telephone connection is sufficient to continue the call. If so, processing returns to step 66 until either the quality degrades or the call is finished. If at step 68 it is determined that the quality is insufficient to carry the call, then the source PBX can reroute the telephone call to the intended recipient on the PSTN at a step 70 without user intervention.

[0018] As can be seen from the above description, the present invention is a communication system having alternative paths on which a call can be routed. By determining whether an intended recipient has the ability to transmit data on an alternative path such as the Internet, the alternative path can be used instead of a traditional PSTN. Telephone calls placed on alternative paths avoid the charges that are incurred each time a call is transmitted on the PSTN.

[0019] Although the present invention has been described with respect to the preferred embodiment, it will be appreciated by those skilled in the art that changes can be made. For example, it is possible that the telephony Internet servers could be located at a central telephone office for users that are not connected to the PSTN through a private branch exchange. The central office would query whether a central office that serves the intended recipient is connected with a telephony Internet server and, based on the answer, could route a telephone call either on the Internet or on the PSTN.

Claims

The embodiments of the invention in which an exclusive property or privilege is claimed are defined as follows:

1. A method of muting a telephone call over a public switched telephone network or a global wide area computer network, comprising:

    receiving a telephone number that is associated with a desired recipient of the telephone call;
    setting up a telephone call to the desired recip-

ient on the public switched telephone network;

determining whether the recipient has a capability to receive the telephone call on the global wide area computer network before the telephone call is connected a the public switched telephone network; and

connecting the telephone call on the global wide area computer network if the recipient has the capability to receive the telephone call on the global wide area network.

2. The method of Claim 1, wherein the telephone call is transmitted on an ISDN line over the public switched telephone network and wherein the step of determining whether the recipient has the capability to receive the telephone on the global wide area computer network comprises:

transmitting a signaling packet to the desired recipient that indicates that a source of the telephone call is equipped with a telephony Internet server, and

receiving from the desired recipient a signaling packet that indicates that the desired recipient is equipped with a telephony internet server.

3. The method of Claim 2, further comprises:

monitoring a time period before a signaling packet is received from the desired recipient that indicates that the desired recipient is equipped with a telephony internet server; and

comparing the time period with a predetermined time threshold and if the time period exceeds the time threshold, then completing the telephone call set up on the public switched telephone network.

4. The method of Claim 3, wherein the time threshold is varied depending on the source of the telephone call.

5. The method of Claim 3, further comprising:

determining if a signaling packet is received from the desired recipient that indicates the desired recipient is equipped with a telephony Internet server after the telephone call has been set up on the public switched telephone network, and if so, connecting the telephone call on the global wide area computer network at the and of a minimum billing increment of the public switched telephone network.

6. The method of Claim 1, further comprising monitoring the quality of the telephone call on the global wide area computer network; and

transfering the telephone call to the public switched telephone network if the quality is degraded.

7. The method of Claim 6, wherein the telephone call transmitted as a number of data packets on the global wide area network and wherein the step of monitoring the quality of the telephone call on the global wide area network, comprises:

determining the rate at which data packets are sent by and received by the telephony Internet servers; and

determining if the rate of the data packets is less than a rate threshold and if so, declaring the quality of the telephone call to be degraded.

8. The method of Claim 6, wherein the telephone call is transmitted as a number of data packets on the global wide area network and wherein the step of monitoring the quality of the telephone call on the global wide area network comprises:

determining a delay time required before a data packet is transmitted on the global wide area network; and

determining if the delay time is greater than a delay time threshold and if so declaring the quality of the telephone call to be degraded.

9. The method of Claim 6, wherein the telephone call is transmitted as a number of data packets on the global wide area network and wherein the step of monitoring the quality of the telephone call on the global wide area network comprises:

determining a rate at which data packets are transmitted and received by the telephony Internet servers and a delay time required before a data packet is transmitted on the global wide area network;

comparing the rate and delay time to a rate and delay time threshold and if the rate is less than the rate threshold, or if the delay time is greater than the delay threshold, declaring the quality of the telephone call to be degraded.

10. A communication system for muting a telephone call over a public switched telephone network or over a global wide area computer network, comprising:

a private branch exchange that connects calls between one or more source telephones and the public switched telephone network;

a telephony Internet server, coupled to the private branch exchange that can route a telephone call over the global wide area network, wherein the private branch exchange operates

to place a call to a second private branch exchange that connects the telephone call to an intended recipient network and to determine whether the second private branch exchange is coupled to a telephony Internet server and if so, routes the telephone call to the intended recipient on the global wide area computer network.

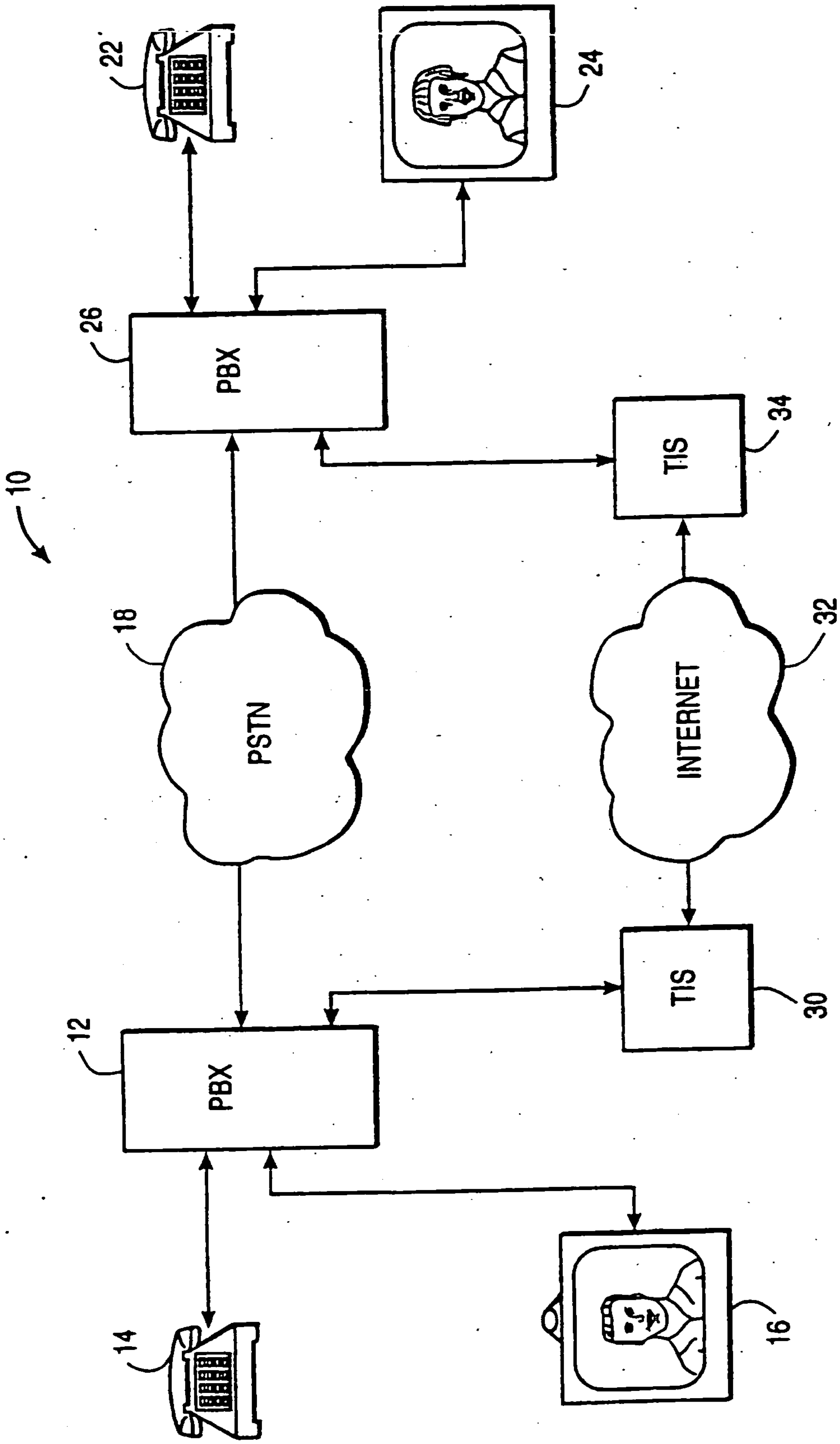

FIG_1

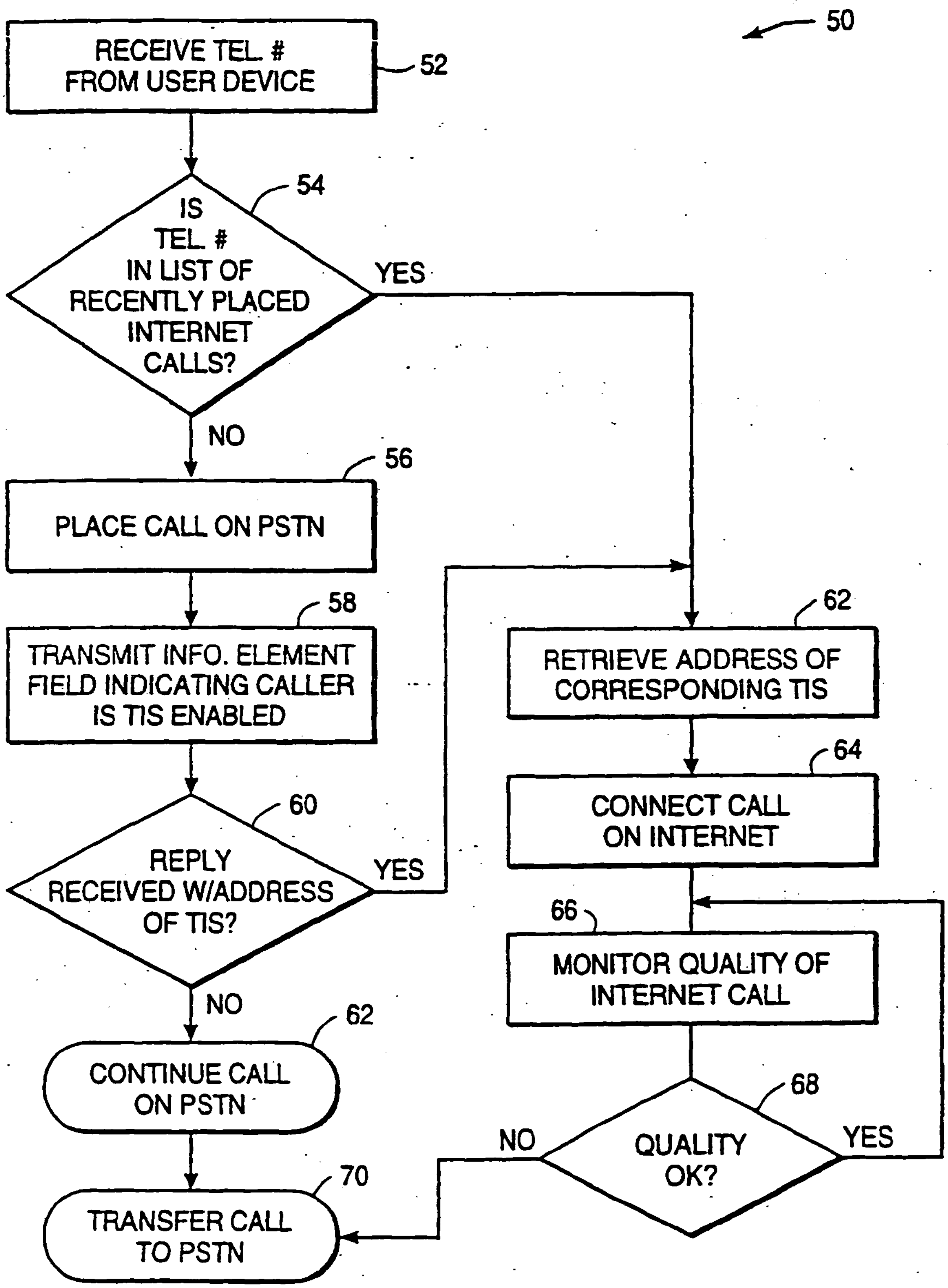
50
RECEIVE TEL. # FROM USER DEVICE
52
IS TEL. # IN LIST OF RECENTLY PLACED INTERNET CALLS?
54
YES
NO
PLACE CALL ON PSTN
56
TRANSMIT INFO. ELEMENT FIELD INDICATING CALLER IS TIS ENABLED
58
REPLY RECEIVED W/ADDRESS OF TIS?
60
YES
NO
CONTINUE CALL ON PSTN
62
TRANSFER CALL TO PSTN
70
RETRIEVE ADDRESS OF CORRESPONDING TIS
62
CONNECT CALL ON INTERNET
64
66
MONITOR QUALITY OF INTERNET CALL
QUALITY OK?
68
NO
YES

FIG_2

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(71) Applicant: **Siemens Information and Communication Networks, Inc.**
Boca Raton, Florida 33487 (US)

(72) Inventors:
- **Shaffer, Shmuel**
  Palo Alto, CA 94301 (US)
- **Beyda, William J.**
  Cupertino, CA 95014 (US)

(74) Representative: **Allen, Derek**
Siemens Group Services Limited,
Intellectual Property Department,
Siemens House,
Oldbury
Bracknell, Berkshire RG12 8FZ (GB)

(54) **Alternative telephone call routing system**

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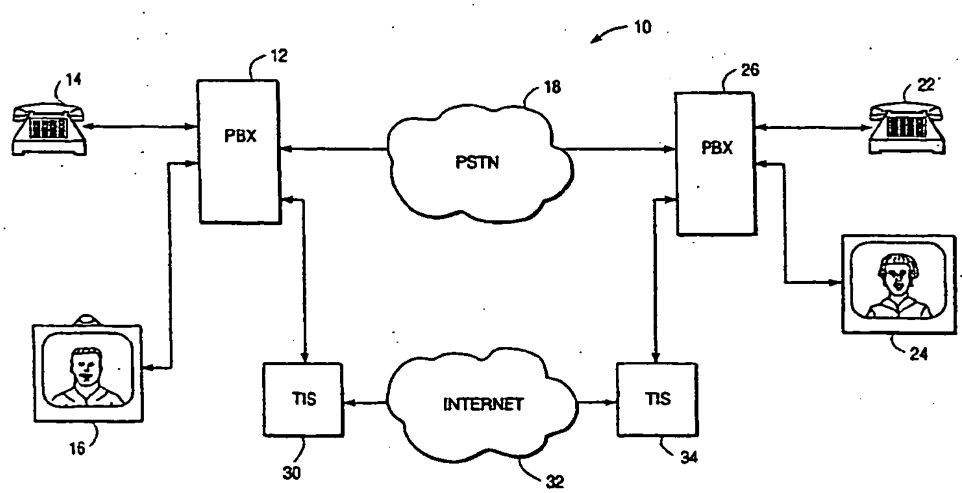

FIG_1

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European Patent Office

EUROPEAN SEARCH REPORT

Application Number
EP 98 11 8543

DOCUMENTS CONSIDERED TO BE RELEVANT			
Category	Citation of document with indication, where appropriate, of relevant passages	Relevant to claim	CLASSIFICATION OF THE APPLICATION (Int.Cl.6)
P,X	WO 98 11704 A (DIALNET INC) 19 March 1998 (1998-03-19) * page 12, line 24 - page 13, line 24 *	1	H04M7/00 H04M7/12 H04Q3/62 H04L29/06
A	EP 0 732 835 A (AT & T CORP) 18 September 1996 (1996-09-18)		
A	HOUGHTON T F ET AL: "A PACKET TELEPHONY GATEWAY FOR PUBLIC NETWORK OPERATORS" ISS '97. WORLD TELECOMMUNICATIONS CONGRESS. (INTERNATIONAL SWITCHING SYMPOSIUM). GLOBAL NETWORK EVOLUTION: CONVERGENCE OR COLLISION? TORONTO, SEPT. 21 - 26, 1997. ISS. WORLD TELECOMMUNICATIONS CONGRESS. (INTERNATIONAL SWITCHING SYMPOSIUM), TORONTO, P, vol. 2, 21 September 1997 (1997-09-21), pages 35-44, XP000704453		
A	ELLIOTT K: "GATEWAYS TO THE TELEPHONE CLOUD" TELECOMMUNICATIONS, DEDHAM, MA, US, 1 April 1997 (1997-04-01), pages 41-42, XP000770891 ISSN: 0040-2494		TECHNICAL FIELDS SEARCHED (Int.Cl.6) H04M H04Q

The present search report has been drawn up for all claims

Place of search	Date of completion of the search	Examiner
THE HAGUE	17 June 2002	Vandevenne, M

CATEGORY OF CITED DOCUMENTS

X : particularly relevant if taken alone
Y : particularly relevant if combined with another document of the same category
A : technological background
O : non-written disclosure
P : intermediate document

T : theory or principle underlying the invention
E : earlier patent document, but published on, or after the filing date
D : document cited in the application
L : document cited for other reasons

& : member of the same patent family, corresponding document

EPO FORM 1503 03.82 (P04C01)

ANNEX TO THE EUROPEAN SEARCH REPORT ON EUROPEAN PATENT APPLICATION NO. EP 98 11 8543

This annex lists the patent family members relating to the patent documents cited in the above-mentioned European search report.
The members are as contained in the European Patent Office EDP file on
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17-06-2002

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			US	5995606	A	30-11-1999

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For more details about this annex : see Official Journal of the European Patent Office, No. 12/82

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(71) Applicant: Nortel Networks Corporation
Montreal, Quebec H2Y 3Y4 (CA)

(72) Inventor: Garvin, Eugene Declan
Galway City (IE)

(74) Representative: Bewley, Ewan Stuart
Nortel Networks
Intellectual Property Law Group
London Road
Harlow, Essex CM17 9NA (GB)

(54) **Data network call handling method**

(57) A voice call is carried between a calling party and a called party over a data network. During the call, the calling party or the called party can instruct the system to select an alternative routing for the call, and the system transfers, in response to the instruction, the call to the alternative routing. The alternative routing can be a switched network (PSTN) routing for the call. The alternative routing can be established before dropping the voice over data network call, such as by conferencing the new call, to provide a seamless handover. The instruction from the calling parties can be made by a key sequence.

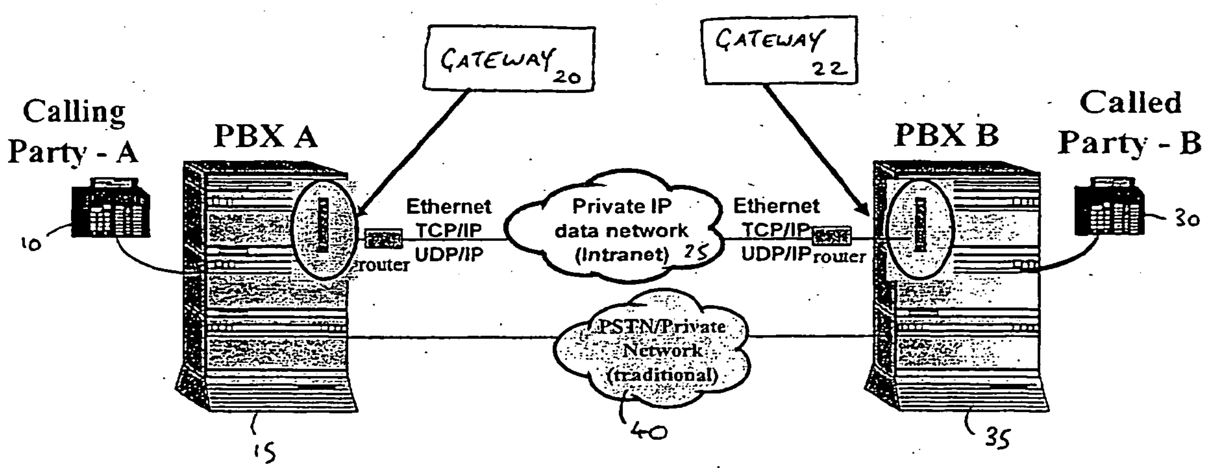


Fig. 2

EP 1 014 667 A2

Printed by Jouve, 75001 PARIS (FR)

Description

TECHNICAL FIELD

[0001] This invention relates to a method and apparatus for handling a communications call and in particular relates to the handling of a voice call over a data network.

BACKGROUND OF THE INVENTION

[0002] There is an increasing interest in providing voice communications over data networks. One of the most common ways for handling voice calls over a data network is voice over internet (VoIP), using internet protocol (IP) techniques. The use of data networks to carry voice traffic can significantly reduce costs. However, there are some disadvantages in using data networks to carry voice traffic. Data networks have less stringent delay criteria than dedicated voice networks, which can lead to noticeable, distracting delays during conversations. Following the establishment of an end-to-end VoIP call between two parties, network congestion may occur which results in increased packet loss and/or latency through the IP network.

[0003] It is known for the network to route voice over data network calls via the PSTN in the event that quality of a call falls below predefined thresholds. However, this may still result in an overall call which is of unacceptable quality to the calling parties.

[0004] The present invention seeks to provide an alternative way of handling a call.

SUMMARY OF THE INVENTION

[0005] A first aspect of the present invention provides a method of handling a call between a calling party and a called party, the call being a voice over data network call, the method comprising the steps of:

- receiving at a switching entity, during the call, an instruction from one of the calling party and the called party to select an alternative routing for the call; and,
- transferring, in response to the instruction, the call to the alternative routing.

[0006] This has advantage of allowing users to obtain an improved quality of service in situations where network performance is inconsistent. Allowing the calling parties themselves to determine an acceptable quality of service (QoS) for the call and to take alternative action can help encourage users who are sceptical of the performance of VoIP and are unwilling to commit to a VoIP based network without a good fallback option.

[0007] Preferably, the alternative routing is a switched telephone network routing for the call, which should offer a higher quality of service than the data network.

[0008] Preferably the step of transferring the call includes the steps of: initiating a new call via the alternative routing and dropping the existing call after the new call has been established. This has the advantage of presenting a seamless handover, with minimal disruption to the calling parties. Advantageously the step of transferring the call includes the step of conferencing the new call into the existing call before dropping the existing call.

[0009] Where there is a calling party switch associated with the calling party and a called party switch associated with the called party, both switches having an interface to the alternative routing, the step of conferencing can include a step of conferencing a port of the alternative routing interface of the calling party switch before initiating a call via the alternative routing. This allows a connection to remain between the calling parties even in the event that the data network call should fail before fallback to the alternative routing has been completed.

[0010] Advantageously the alternative routing is made via a dedicated port on the called party switch. By using a dedicated port, the called party switch can realise that the call is one which requires fallback treatment. Advantageously the step of conferencing includes the calling party switch signalling the identity of the called party to the called party switch.

[0011] Preferably, the step of receiving comprises receiving a signal which identifies the alternative routing. The, or each, alternative routing for the call can be identified by a particular message. The user can select the alternative routing by dialling a key sequence or by pressing a pre-programmed key on the terminal.

[0012] The signalling can be by a dual tone multifrequency (DTMF) signal or some other signalling message, such as an ISDN message.

[0013] Another aspect of the invention provides apparatus for use in a communications system which is handling a call between a calling party and a called party, the call being a voice over data network call, the apparatus comprising:

- means for receiving, during the call, an instruction from one of the calling party and the called party to select an alternative routing for the call; and,
- means for transferring, in response to the instruction, the call to the alternative routing.

[0014] This apparatus may be incorporated in a telecommunications switch.

[0015] Preferred features may be combined as appropriate, and may be combined with any of the aspects of the invention, as would be apparent to a person skilled in the art.

BRIEF DESCRIPTION OF THE DRAWINGS

[0016] For a better understanding of the invention,

and to show by way of example how it may be carried into effect, embodiments will now be described with reference to the accompanying drawings, in which:

Figure 1 shows a typical system which allows a telephony call to be made via a data network;
Figure 2 shows a system which provides a fallback path via an alternative routing;
Figure 3 shows message flows for the system of figure 2;
Figure 4 shows signalling to achieve the fallback method: and
Figure 5 shows one of the PBXs in more detail.

DESCRIPTION OF PREFERRED EMBODIMENTS

[0017] Figure 1 shows a typical system which allows a telephony call to be made via a data network. Terminals of a calling party 10 and a called party 30 are each linked to a respective private branch exchange (PBX) 15, 35. The PBXs 15. 35 each have an interface to a gateway device 20. 22 which performs conversion of traffic between the format used in the PBX and the format necessary for transport over the data network 25. The gateway or a gatekeeper entity handles translation between dialled number and IP addresses where this is required. The traffic is typically carried according to TCP/IP or UDP/IP formats over data network 25: H.323 typically being carried by TCP packets and speech data by UDP packets The speech or other voice-band information is packaged into data packets which each have a header that carries information to allow the packet to be routed across data network 25. The data network 25 includes routers 21, 26 which perform the routing across the data network 25. The above apparatus is known in the art. The data network is typically a private IP data network or intranet, which is designed and managed to offer a good quality of service. Usually, the delay experienced by data packets crossing the network will be low enough to offer a quality of service which is sufficient, in most circumstances, to support voice traffic.

[0018] Figure 2 shows the system of figure 1 with the addition of a switched telephone network link 40 between PBXs 15, 25. The switched network link can be via the public switched telephone network (PSTN) or a private network of leased lines. PBXs 15, 25 include suitable interfaces to the switched network, and these would usually already be part of the PBXs.

[0019] The fallback process will now be described with reference to figure 3 (which shows the system of figure 2 with the addition of message flows) and figure 4. The fallback process allows the party who initiated a voice over IP call to fallback to a PSTN based call by dialling an appropriate key sequence on their dialpad. The sequence causes a separate PSTN call to be established to the destination site. The call is terminated at the remote site and the calls are merged to provide a seamless handover. The original VoIP call can then be dropped. The calling party (or the called party) can make the judgement on whether to fallback or not and can invoke the fallback without interrupting the call. The detail of the implementation is dependent upon the host PBX and gateway used.

[0020] A typical call scenario will now be described in more detail.

Call Scenario

[0021]

1. Calling party A 10 dials the called party B 30. Called party B answers, thereby establishing an end to end VoIP call (step 100, fig.4.) The call is set up as a voice over data network call via PBX A, gateway 20, data network 25, gateway 22 and PBX B. Both parties A and B are connected to a PBX and have a signalling path to their respective PBX. The voice path may be end-to-end IP, or the parties may be communicating via VoIP trunk gateways on either end, as shown in fig. 3.

2. At some point during the call, network congestion occurs and calling party A decides that the quality of the speech path is unacceptable and fallback is required. Party A then dials a predetermined key sequence, e.g. '*777', (step 101, fig.4.) The signalling can be DTMF signalling. This signalling is detected by PBX A, which recognises the sequence as a fallback request for this call. A signalling detection resource is maintained online, in 'listening mode', for the duration of the call to detect the DTMF sequence. Alternatively, for PBX proprietary telephone sets, a specific feature key can be defined on the user's terminal that performs the same function as dialing the '*777' code. Upon receipt of the appropriate sequence, the call control software will commence the setup of the alternative call. This call can be setup between dedicated fallback units on the two systems (PBXs) that are party to the call or alternatively, can use non-dedicated ports provided the necessary signalling information can be exchanged to indicate to the far side that this is a fallback call. Either way, the system that is terminating the alternative call should know (i) that this is a fallback call and, (ii) the terminating party for the call.

3. An additional, optional, step 102 can be invoked at this point. A conference can be set up between the originating party on PBX A and originating fallback port on PBX A prior to initiating the fallback PSTN call. This has the effect of allowing the fallback call to complete once PBX B has replied, even if the VoIP call subsequently drops.

4. PBX A initiates a call to PBX B (step 103) using the PSTN 40, or some alternative route that will by-

pass the original IP call path. The connection can be a TIE or DID. A TIE call is made over a dedicated leased line connection (analog/digital) between two PBXs, and while the PSTN infrastructure may be used to route the call, the call is not terminated on the PSTN. A DID (Direct Inward Dialled) call is terminated on the PSTN and contains sufficient information to terminate directly on the called party's extension. PBX A dials the number of a dedicated fallback port on PBX B.

5. PBX B answers on it's fallback port and immediately switches in a DTMF detector. Once PBX B receives the call, it must then receive the digit information regarding who the terminating party is. (This can be via in-band tones that are detected by a tone detection resource on PBX B.) PBX B then establishes a conference (step 105) between the terminating party and the terminating fallback port, using a conference resource.
Note: If the fallback call is an ISDN call, the exchange of Party B's DN can be communicated via a FACILITY message rather than signalling tones.

6. PBX A detects the answer from PBX B and transfers Party B's directory number (DN) as a series of DTMF digits to PBX B (step 104). As an alternative to signalling the DN of party B, PBX A can identify the on-going VoIP connection.

7. Party B's DN digits are detected by the fallback port on PBX B. PBX B initiates a conference with Party B's DN (step 104). Once this is established successfully, it then signals back to the originating system that the call has been successfully conferenced at the far end (step 110) using an appropriate signal. As soon as PBX A receives the completion signal, it can immediately transfer the calling party to the fallback port (step 115), dropping the VoIP gateway call in the process. If a conference at PBX A was established at step 102 then the conference is ended at this point by dropping the VoIP call. This establishes an end to end connection to the terminating party on PBX B. As soon as PBX B detects the dropping of the VoIP gateway call, the conference will be converted to a simple call, freeing up the VoIP gateway port and the conference resource (step 120).

8. Party A is now left talking to Party B via a PSTN (or alternative) connection and the fallback is complete (step 108.) The call is now equivalent to if the caller had dialled a PSTN based trunk route.

[0022] It is possible that the IP link can fail before the fallback to the PSTN is complete. If conferencing is used on PBX A during call setup and PBX B has answered, the call will be seamlessly completed even though the VoIP call subsequently drops. If the VoIP call drops before PBX B answers, the call will be presented to the terminating party of PBX B as a new (PSTN) call.

[0023] The above implementation requires the use of a hardware or software entity on PBX B receiving the call that is capable of terminating the fallback call via the PSTN and initiating the required transfer. A Nortel Networks MERIDIAN™ VPS card can perform this function.

[0024] Figure 5 shows a high level view of a typical circuit or packet switched based PBX. The gateways in each case also contain the actual physical interface to the particular network specified. For the VoIP Gateway, the interface is usually Ethernet.

[0025] The tone detection resource 60 is shown as a separate entity. It can co-reside with the VoIP Gateway 20 and be dedicated to the gateway or it can be a system-wide resource available to other gateways 80, 90 in the system.

[0026] The call control software 50 is responsible for call routing within the system in response to messages from the gateways, the tone detection resource and the terminal interfaces. In the case of user-initiated fallback, the call control software will respond to the fallback digit sequence received via the terminal interface 95, detected by the tone detector 60 or by the gateway itself (where the gateway has built-in tone detection facilities).

Alternatives

[0027] Where the called party initiates the fallback, there are two requirements: (i) the called party must know exactly the calling party's number, and (ii) the called party must be configured to support this functionality and have a tone detection resource online to detect the fallback sequence. If both of these conditions are satisfied, the call sequence is simply a reverse of that previously described.

[0028] The port terminating the fallback call on side B could be a DN controlled by a 3rd party application that is responsible for the call control or it could also be implemented as a telset emulation port on a card such as the Nortel Networks MERIDIAN™ VPS card. This is capable of emulating MERIDIAN™ proprietary terminals and is capable of delivering the required functionality to answer the call, detect digits and then transfer the call with/without an external CTI interface. The VPS card contains both simple conferencing and tone detection facilities. For a call in fallback mode, a port on the VPS card is conferenced in at the time that this fallback route is established, by the call control software. While in non-fallback mode, this port operates in 'listen only' mode, checking for the '*777' digit sequence. Once the sequence is detected, the VPS card has the ability to initiate a call to an outgoing trunk port, transfer across the DTMF digit string and then signal to the call control software once the call is established. The call control software then tears down the VoIP call and a call path exists

between the calling party, the VPS card and the outgoing trunk (which doesn't need to be dedicated).

[0029] While the above description shows an alternative routing via the PSTN, it will be appreciated that other alternative routes could be used, such as frame relay or ATM, providing that these do not suffer from the same impairments as the network which carries the IP packets.

[0030] While the above description shows how the alternative routing may be applied between a calling party switch PBX A and a called party switch PBX B, it will be appreciated the alternative routing could also be used for a shorter leg of the total path between the calling party and called party, such as between two switches which are not directly connected to the calling and called parties. The description of the switches as being "associated with the calling party" and "associated with the called party" are intended to mean "associated with the calling party end of the path between the calling and called parties" and "associated with the called party end of the path between the calling and called parties."

[0031] The above describes how a predetermined key sequence or a programmed key on the user's terminal can be used to instruct the switch to perform a transfer to an alternative routing. There can be a plurality of different key sequences/programmed keys, each representing a different alternative routing for the call.

Claims

1. A method of handling a call between a calling party and a called party, the call being a voice over data network call, the method comprising the steps of:

   - receiving at a switching entity, during the call, an instruction from one of the calling party and the called party to select an alternative routing for the call; and,
   - transferring, in response to the instruction, the call to the alternative routing.

2. The method according to claim 1 wherein the alternative routing is a switched telephone network routing for the call.

3. The method according to claim 1 wherein the step of transferring the call includes the steps of: initiating a new call via the alternative routing and dropping the existing call after the new call has been established.

4. The method according to claim 3 wherein the step of transferring the call includes the step of conferencing the new call into the existing call before dropping the existing call.

5. The method according to claim 4 wherein there is a calling party switch associated with the calling party and a called party switch associated with the called party, both switches having an interface to the alternative routing and wherein the step of conferencing includes the calling party switch signalling the identity of the called party to the called party switch.

6. The method according to claim 4 wherein there is a calling party switch associated with the calling party and a called party switch associated with the called party, both switches having an interface to the alternative network and wherein the step of conferencing includes conferencing a port of the alternative routing interface of the calling party switch before initiating a call via the alternative routing.

7. The method according to claim 1 wherein there is a calling party switch associated with the calling party and a called party switch associated with the called party, both switches having an interface to the alternative routing, and the alternative routing is made via a dedicated port on the called party switch.

8. The method according to claim 1 wherein the step of receiving comprises receiving a signal which identifies the alternative routing.

9. The method according to claim 1 wherein the step of receiving comprises receiving a signalling message.

10. The method according to claim 9 wherein the step of receiving comprises receiving a signalling message which is one of a DTMF signal and an ISDN message.

11. A method of operating a switch in a communications system which is handling a call between a calling party and a called party, the call being a voice over data network call, the method comprising the steps of:

    - receiving, during the call, an instruction from one of the calling party and the called party to select an alternative routing for the call; and,
    - transferring, in response to the instruction, the call to the alternative routing.

12. Apparatus for use in a communications system which is handling a call between a calling party and a called party, the call being a voice over data network call, the apparatus comprising:

    - means for receiving, during the call, an instruction from one of the calling party and the called party to select an alternative routing for the call; and,
    - means for transferring, in response to the in-

struction, the call to the alternative routing.

13. A telecommunications switch incorporating the apparatus according to claim 12.

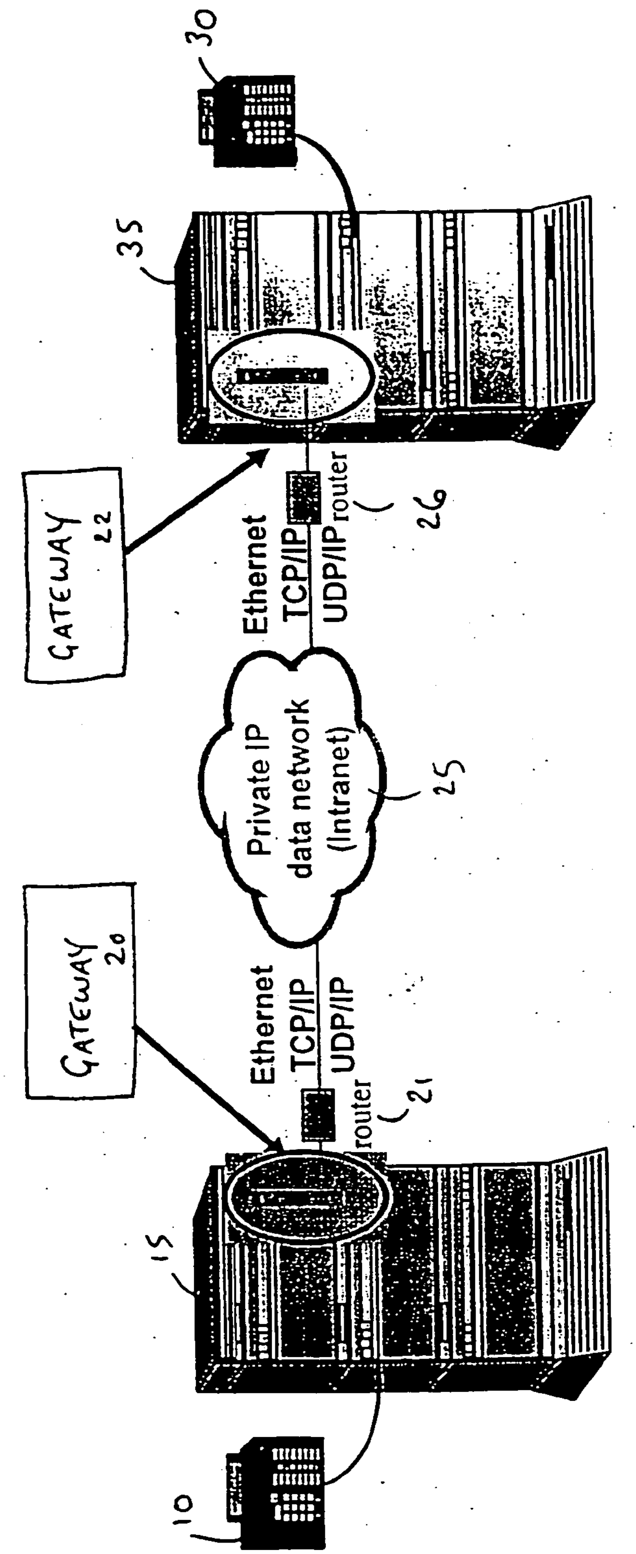
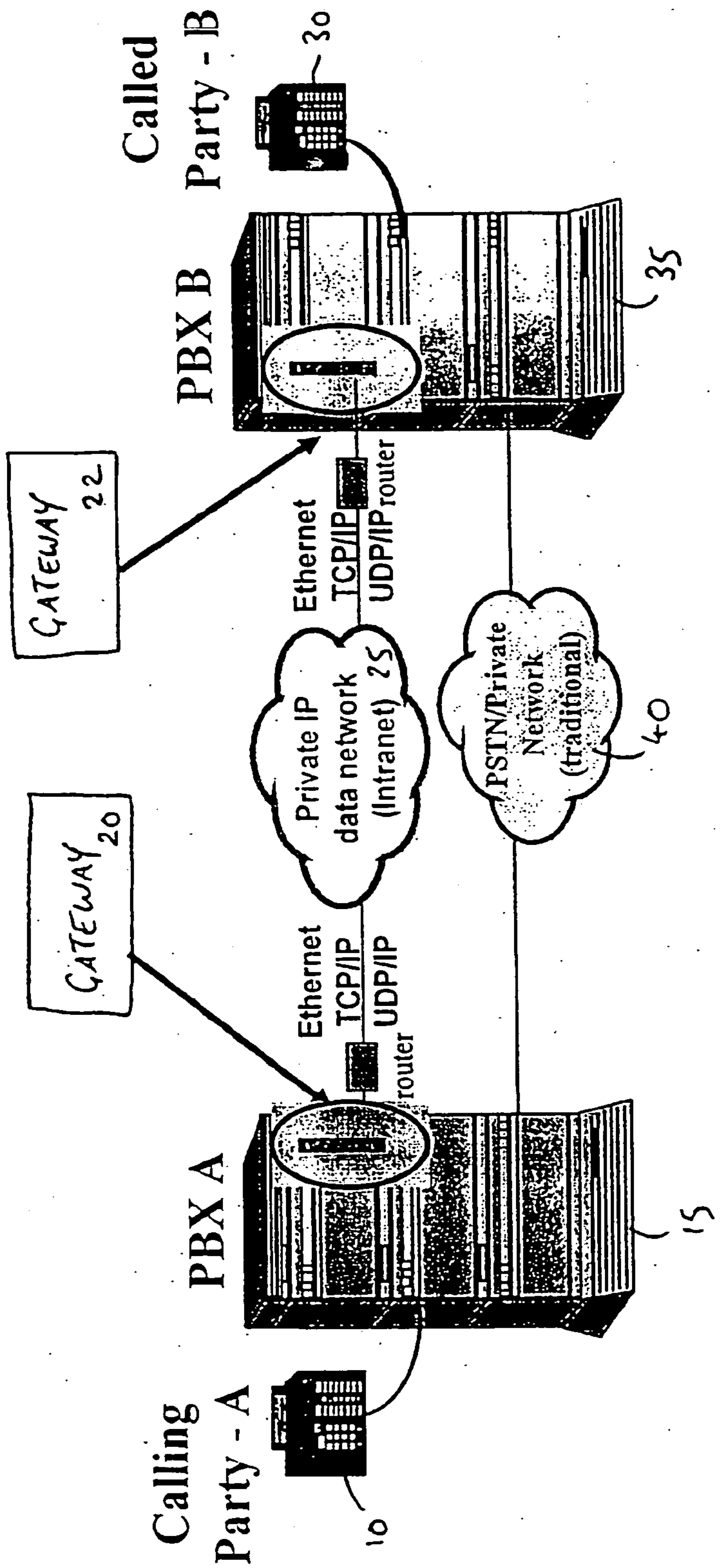


FIG. 1 (PRIOR ART)



Calling Party - A
10
PBX A
15
GATEWAY 20
Ethernet
TCP/IP
UDP/IP
router
Private IP data network (Intranet) 25
PSTN/Private Network (traditional)
40
GATEWAY 22
Ethernet
TCP/IP
UDP/IP
router
PBX B
35
Called Party - B
30

FIG. 2

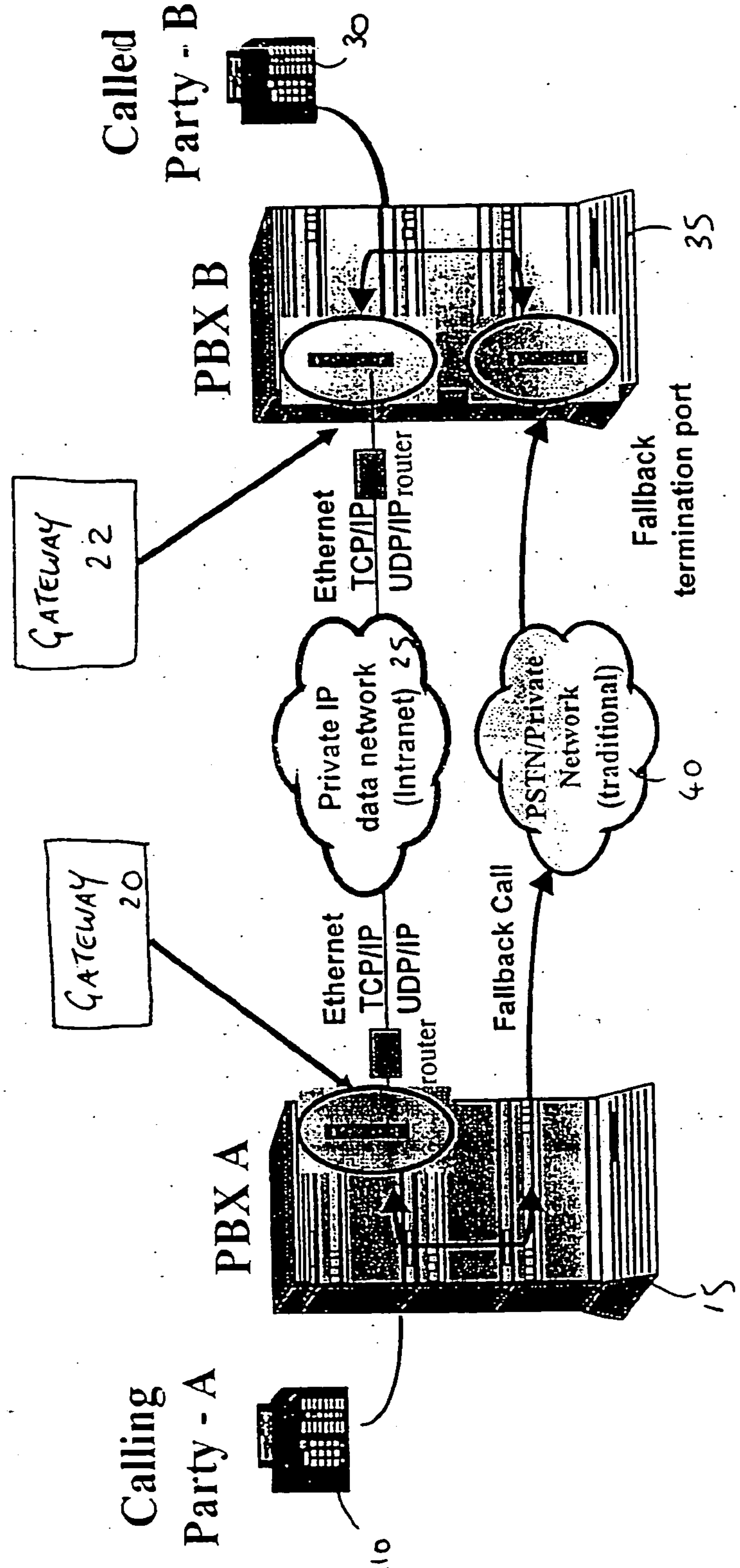

Calling Party - A
10
PBX A
15
GATEWAY 20
Ethernet
TCP/IP
UDP/IP
router
Fallback Call
Private IP data network (Intranet)
25
PSTN/Private Network (traditional)
40
GATEWAY 22
Ethernet
TCP/IP
UDP/IP
router
Fallback termination port
PBX B
35
Called Party - B
30

FIG. 3

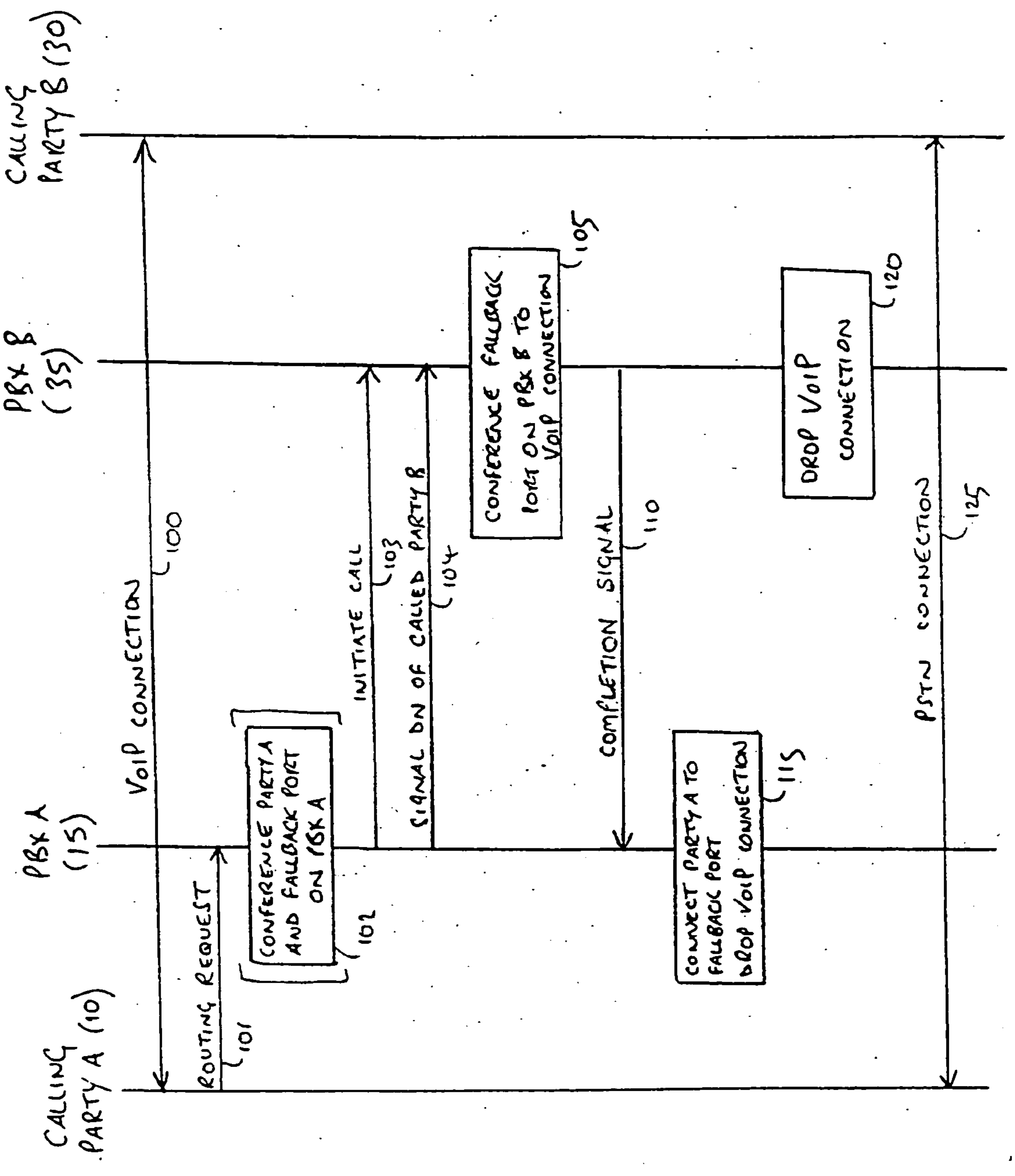

CALLING PARTY A (10)
PBX A (15)
PBX B (35)
CALLING PARTY B (30)
VoIP CONNECTION
100
ROUTING REQUEST
101
CONFERENCE PARTY A AND FALLBACK PORT ON PBX A
102
INITIATE CALL
103
SIGNAL DN OF CALLED PARTY B
104
CONFERENCE FALLBACK PORT ON PBX B TO VoIP CONNECTION
105
COMPLETION SIGNAL
110
CONNECT PARTY A TO FALLBACK PORT DROP VoIP CONNECTION
115
DROP VoIP CONNECTION
120
PSTN CONNECTION
125

FIG. 4

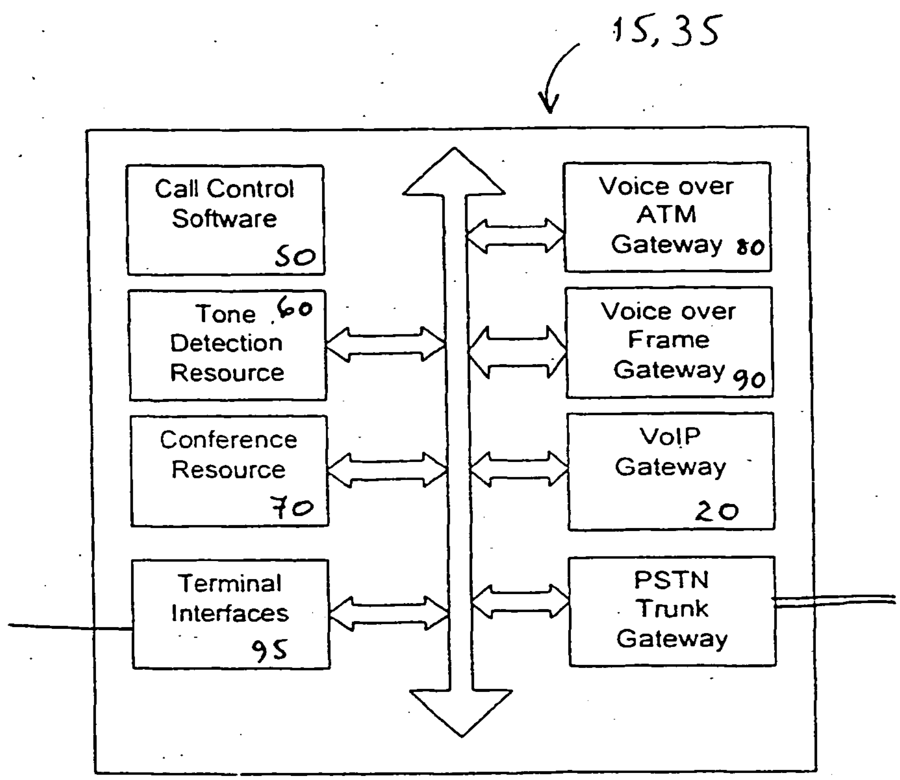

15, 35
Call Control Software 50
Tone Detection Resource 60
Conference Resource 70
Terminal Interfaces 95
Voice over ATM Gateway 80
Voice over Frame Gateway 90
VoIP Gateway 20
PSTN Trunk Gateway

FIG. 5

(19) Europäisches Patentamt
European Patent Office
Office européen des brevets

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(71) Applicant: **Nortel Networks Limited**
Montreal, Quebec H2Y 3Y4 (CA)

(72) Inventor: **Garvin, Eugene Declan**
Galway City (IE)

(74) Representative: **Bewley, Ewan Stuart**
Nortel Networks
Intellectual Property Law Group
London Road
Harlow, Essex CM17 9NA (GB)

(54) **Data network call handling method**

(57) A voice call is carried between a calling party and a called party over a data network. During the call, the calling party or the called party can instruct the system to select an alternative routing for the call, and the system transfers, in response to the instruction, the call to the alternative routing. The alternative routing can be a switched network (PSTN) routing for the call. The alternative routing can be established before dropping the voice over data network call, such as by conferencing the new call, to provide a seamless handover. The instruction from the calling parties can be made by a key sequence.

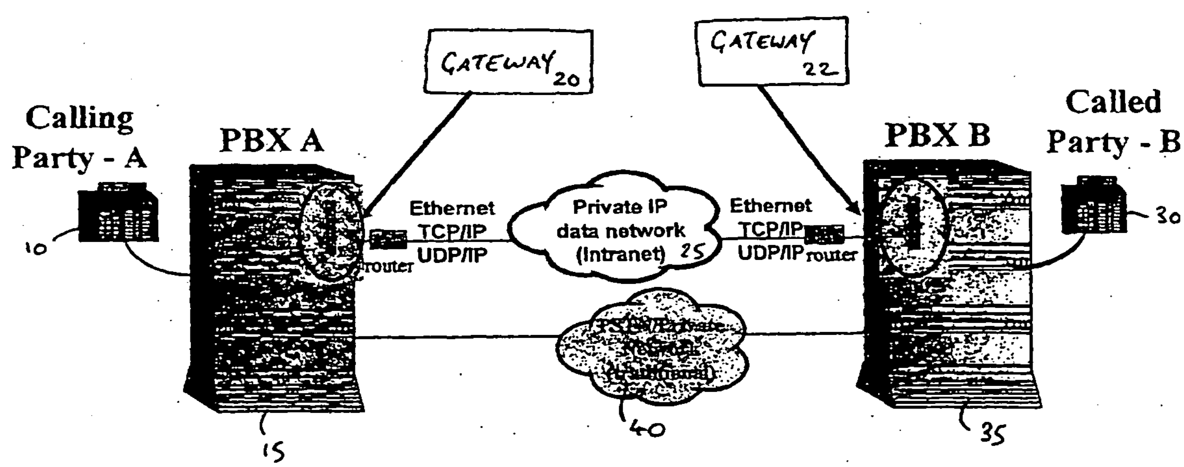

FIG. 2

EP 1 014 667 A3

European Patent Office

EUROPEAN SEARCH REPORT

Application Number
EP 99 31 0002

DOCUMENTS CONSIDERED TO BE RELEVANT

Category	Citation of document with indication, where appropriate, of relevant passages	Relevant to claim	CLASSIFICATION OF THE APPLICATION (Int.Cl.7)
X	DE 196 45 368 A (TELES AG) 16 April 1998 (1998-04-16)	1-13	H04M7/00
X	* page 4, line 41-56 *	1,11-13	
X	* column 4, line 53-56 *	2	
X	* column 10, line 1-9 *	3	
X	* column 12, line 37 - column 13, line 32 *	8	
X	* column 9, line 59-68 *	9	
X	* column 4, line 41-48 *	10	
	* column 9, line 59-68 *		

TECHNICAL FIELDS SEARCHED (Int.Cl.7)

H04M

The present search report has been drawn up for all claims

Place of search	Date of completion of the search	Examiner
THE HAGUE	24 May 2002	Cremer, J

CATEGORY OF CITED DOCUMENTS

X : particularly relevant if taken alone
Y : particularly relevant if combined with another document of the same category
A : technological background
O : non-written disclosure
P : intermediate document

T : theory or principle underlying the invention
E : earlier patent document, but published on, or after the filing date
D : document cited in the application
L : document cited for other reasons

& : member of the same patent family, corresponding document

EPO FORM 1503 03.82 (P04C01)

ANNEX TO THE EUROPEAN SEARCH REPORT ON EUROPEAN PATENT APPLICATION NO. EP 99 31 0002

This annex lists the patent family members relating to the patent documents cited in the above-mentioned European search report.
The members are as contained in the European Patent Office EDP file on
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24-05-2002

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		AT 186167 T	15-11-1999
		WO 9815933 A2	16-04-1998
		DE 59700636 D1	02-12-1999
		EP 0929884 A2	21-07-1999
		ES 2140965 T3	01-03-2000
		NO 991625 A	03-06-1999

EPO FORM P0459

For more details about this annex : see Official Journal of the European Patent Office, No. 12/82

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(11) EP 0 920 176 A2

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(71) Applicant: NORTHERN TELECOM LIMITED
Montreal, Quebec H2Y 3Y4 (CA)

(72) Inventor: Cruickshank, Brian
Oakville, Ontario L6L 4N8 (CA)

(74) Representative: Ryan, John Peter William
Nortel Networks
Intellectual Property Law Group
London Road
Harlow, Essex CM17 9NA (GB)

(54) Automatic backup trunking for voice over the internet

(57) To reduce telephone toll costs to a user, a PBX preferentially establishes a call to a destination number (DN) over a WAN or the internet. The PBX determines the available connection types available by querying look-up tables for the particular DN. If no alternatives to the PSTN are available, the call is routed over the PSTN. Where a WAN or internet connection is available, the call is then routed over this alternative service. If the Quality of Service (QoS) over the computer network connection falls below a specified threshold, a second parallel connection is made over the PSTN and the call is then transferred to the PSTN. The user is notified of this change in service. During the PSTN connection, the PBX polls the alternative service and, upon the QoS rising above a specified threshold, the call is then routed back to the alternative service and the PSTN connection is torn down. The user is again notified of this change in service.

EP 0 920 176 A2

Description

BACKGROUND OF THE INVENTION

[0001] This invention relates to a method and apparatus for routing a telephone call.

[0002] Voice communication over the internet is known. However, such communication typically requires that both parties be logged on to an internet provider, be running compatible voice communication software, and have the necessary hardware (e.g., microphone and speakers). Further, voice communication over the internet may degrade due to congestion.

[0003] The present invention seeks to obviate disadvantages of known voice communication over the internet.

SUMMARY OF THE INVENTION

[0004] According to the present invention, there is provided a method for routing a telephone call, comprising the steps of receiving a destination number (DN) for said call; based on said DN, determining whether a connection is possible through a computer network; where a computer network connection is possible, routing said call through said computer network; where a computer network connection is not possible, routing said call through a switched telephone network.

[0005] According to another aspect of the present invention, there is provided a call router for routing a telephone call, comprising: a receiver for receiving an outgoing call; a detector responsive to said receiver for detecting a destination for said call; a determiner responsive to said destination detector for determining whether or not a connection is possible through a computer network; a route initiator responsive to said determiner for initiating a route for said call through one of said computer network and a switched telephone network.

BRIEF DESCRIPTION OF THE DRAWINGS

[0006] In the figures which show an example embodiment of the invention,

figure 1 is a schematic view of a communication system embodying this invention,
figure 2 is a schematic detail of a portion of figure 1, and
figures 3a and 3b comprise a flow diagram of the program control for a portion of the system of figure 1.

DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENTS

[0007] Turning to figure 1, a communication system illustrated generally at 10 comprises a plurality of telephone appliances 12 connected to a first private branch exchange (PBX) 14 and a plurality of telephone appliances 22 connected to a second PBX 24. PBX 14 and PBX 24 are each connected to a wide area network (WAN), or intranet, 40, an internet 42, and a public switched telephone network (PSTN) 44.

[0008] Referencing figure 2, PBX 14 comprises a processor 52 connected for two-way communication with a memory 54 and having a control input to a switch matrix 56. The switch matrix has inputs 58 from telephone appliances connected to the PBX and outputs 60 to these telephone appliances and to intranet, internet, and PSTN lines. PBX 24 is identically configured.

[0009] The operation of the communication system of figures 1 and 2 is described in conjunction with figures 3a and 3b, which illustrates the program control for processor 52 of PBX 14.

[0010] If a user of telephone appliance 12a goes off-hook and dials a destination number (DN) for telephone appliance 22a, the PBX 14 receives the DN (block 110). The processor of the PBX retrieves a WAN look-up table from memory 54 and determines whether the DN appears in the table (block 112). If yes, this means that the DN is for a telephone appliance associated with a PBX on the WAN and the processor retrieves the WAN address associated with the DN from the look-up table (block 114). The WAN address is an indication of a destination PBX on the WAN to which the called telephone appliance is associated. In the illustrative example, the address for PBX 24 would appear in the WAN table. With this information, PBX 14 routes the call through its switch matrix 56 to a WAN line and initiates routing through the WAN in order to complete the call (block 116), verifying in the process that the Quality of Service (QoS) is high enough to support a real-time telephone conversation by measuring the packet delay.

[0011] If the user of telephone 12a dialled the DN for a telephone appliance not on the WAN or the call required a QoS above that available on the WAN, then PBX 14 next accesses an internet look-up table and searches for the DN (block 118). If an entry is found, the internet address is retrieved (block 120) and the PBX 14 initiates routing through the internet to the destination PBX (block 122), verifying in the process that the QoS is high enough to support a real-time telephone conversation.

[0012] Assuming PBX 14 did not have an internet look-up table entry for the DN, or the QoS was not high enough to support a telephone conversation, the PBX initiates routing of the call through the PSTN (blocks 124, 130).

[0013] Whenever a call is established over the internet, PBX 14 monitors the quality of service (QoS) of the internet call path (block 134). This involves measuring such parameters as packet delay, the number of data packets dropped and throughput. Preferably QoS is measured using the known Real-Time Transport Control Protocol (RTCP). If the QoS falls below a first threshold (block 136), then PBX 14 initiates the setting up of a

parallel call path to the destination PBX over the PSTN. Once this parallel path is established. PBX 14 sends a sequence of in-band tones over the PSTN to the destination PBX which uniquely identifies the internet connection carrying the call's voice path connection (e.g. the calling and called telephone numbers). The destination PBX sends a confirmation tone over the PSTN to PBX 14 indicating when it has found the connection. (This interaction is done over the PSTN instead of over the internet because the internet is assumed to be suffering delays due to congestion at this time.) The confirmation tone is used as a signal for both PBXs to simultaneously switch the voice path from the internet to the PSTN. Since the internet voice path is suffering quality problems such as excessive delay, it will normally be acceptable to switch the voice path without waiting for a silence interval. A notification tone can be sent to the calling and called parties during a silence interval to notify them that the call has been rerouted. Typically a PSTN connection generates higher user charges than an internet connection and so the alerting informs the parties of their use of a higher price connection.

[0014] After the change-over. the internet call path is maintained and PBX 14 sends test packets over the internet call path to allow it to continuously monitor the QoS of the connection (block 144). If the QoS improves so as to exceed a second threshold -- which may be set higher than the first threshold (block 146). PBX 14 monitors for silence on the PSTN connection, then initiates routing of the call through the internet (block 148). The PBX may also send an in-band signal to alert the parties of a switch over back to the internet connection. The PBX then initiates tearing down of the PSTN connection (block 150).

[0015] For the duration of the call, PBX 14 monitors the QoS of the internet connection and re-establishes a PSTN connection whenever necessary.

[0016] In the foregoing, it is assumed that the WAN 40 is able to guarantee a QoS for each connection. If this is not the case, then the PBX 14 monitors and responds to the QoS on the WAN in the same fashion as it monitors and responds to the QoS on the internet.

[0017] By utilising a computer network (intranet or internet) call path in preference to a PSTN call path, the communication system 10 minimizes toll costs of a call. Additionally, the communication system 10 provides a "safety factor" for any call over a computer network in that should the QoS of the call degrade for any reason, the call will be rerouted through the PSTN.

[0018] While figure 1 illustrates two networked PBXs, it will be readily apparent that any number of PBXs may form part of a "corporate" network. When any new PBX is to join the corporate network and this new PBX is connected to the internet, a system operator enters the internet protocol (IP) address of a "reference" PBX in the corporate network. The reference PBX can be any active PBX of the corporate network which has an internet connection. This prompts the new PBX to send a message to this IP address identifying itself as a new PBX on the corporate network along with the range of DNs to which it responds and an authentication code. The reference PBX returns a message which contains a mapping between corporate network DN ranges and IP addresses for all of the PBXs in the corporate network. The new PBX stores this information in a look-up table and then sends a message to each of the PBXs in the corporate network identifying itself as a new PBX on the corporate network and specifying the range of DNs to which it responds. Upon receiving this message, the other PBXs update their look-up table to include this new PBX. This same procedure may be used to incorporate a new PBX in a WAN of the corporate network.

[0019] Efficiency of the corporate network may be further enhanced by a modification wherein each PBX periodically sends test messages to each of the other PBXs in the corporate network to determine the quality of service of the WAN/internet connections between itself and the other PBXs. If it determines that the quality of service with another PBX is not high enough to support an acceptable voice conversation, it will set a "poor Voice Quality" flag in a look-up table indicating that calls to this PBX should be routed over the PSTN. This flag will be cleared when subsequent tests indicate that the quality of service achievable over the WAN/internet connection to this PBX has returned to an acceptable level.

[0020] With this modification, when a user places a call to a remote PBX, the load PBX will look up the IP address of the remote PBX and check the Poor Voice Quality flag associated with that PBX. If the remote PBX has an IP address in the look-up table and its Poor Voice Quality flag is not set, the local PBX will set up the call over the WAN or internet. Otherwise it will set up the call over the PSTN.

[0021] While the illustrative embodiments reference the PSTN, it will be appreciated that this network could equally be a network of leased lines or other switched telephone network. If the switched telephone network does not support an end-to-end digital connection. it may be necessary to convert an incoming call from analog to digital in any known fashion before the call is routed over an internet connection.

[0022] Each PBX in the illustrative embodiment could be replaced by any intelligent switch. Further, instead of programming a PBX or other intelligent switch to perform as described. a special purpose router could be associated with the switch. The switch would then be programmed to query the router for instructions whenever a call arrived and the router would instruct the switch to operate in the manner described for the PBX hereinbefore. As a further alternative, if the switch was a signal switching point (SSP) in an advanced intelligent network (AIN), then, as is standard in an AIN, the SSP queries a supervisory control point (SCP) when a call arrives. The SCP could contain the program control for the SSP such that the SSP operated in the manner hereinbefore described for the PBX.

[0023] In summary, to reduce telephone toll costs to a user, a PBX preferentially establishes a call to a destination number (DN) over a WAN or the internet. The PBX determines the available connection types available by querying look-up tables for the particular DN. If no alternatives to the PSTN are available, the call is routed over the PSTN. Where a WAN or internet connection is available, the call is then routed over this alternative service. If the Quality of Service (QoS) over the computer network connection falls below a specified threshold, a second parallel connection is made over the PSTN and the call is then transferred to the PSTN. The user is notified of this change in service. During the PSTN connection, the PBX polls the alternative service and, upon the QoS rising above a specified threshold, the call is then routed back to the alternative service and the PSTN connection is torn down. The user is again notified of this change in service.

[0024] Other modifications will be apparent to those skilled in the art and, therefore, the invention is defined in the claims.

Claims

1. A method for routing a telephone call, comprising the steps of:

   receiving a destination number (DN) for said call;
   based on said DN, determining whether a connection is possible through a computer network;
   where a computer network connection is possible, routing said call through said computer network;
   where a computer network connection is not possible, routing said call through a switched telephone network.

2. A method claimed in claim 1, including the steps of: where said call is routed through said computer network, monitoring a quality of service for said call and, where said quality of service falls below a threshold, dynamically rerouting said call through said switched telephone network.

3. A method as claimed in claim 2, wherein said step of dynamically rerouting comprises the steps of:

   setting up a call path on said switched telephone network which is parallel to an existing call path for said call on said computer network: and
   switching said call from said computer network path to said switched telephone network path.

4. A method as claimed in claim 2 or claim 3, including the steps of:

   maintaining said computer network path;
   sending test data packets on said computer network path;
   monitoring a quality of service for said test data packets;
   where said test packets quality of service exceeds a threshold, switching said call from said switched telephone network path back to said computer network path and tearing down said switched telephone network path.

5. A method as claimed in any one of claims 2 to 4, including the step of monitoring for a natural break in communication on said computer network path and wherein the step of switching is responsive to said monitoring step.

6. A method as claimed in claim 5, including the step of generating an in-band signal on the switching step.

7. A method as claimed in any preceding claim, wherein said computer network comprises an internet and/or an intranet and wherein said switched telephone network comprises a public switched telephone network (PSTN).

8. A method as claimed in claim 7, including the steps of:

   where an intranet connection is not possible, determining if an internet connection is possible; and
   where an intranet connection is not possible and an internet connection is possible, routing said call through said internet preferentially to routing said call through said switched telephone network.

9. A method as claimed in any preceding claim, including the steps of, where a computer network connection is available:

   repeatedly testing a quality of service of said computer network connection;
   storing an indication of whether or not said quality of service is poor based on said testing;
   determining whether a computer network connection is possible based on said stored indication.

10. A call router for routing a telephone call, comprising:

   a receiver for receiving an outgoing call;
   a detector responsive to said receiver for detecting a destination for said call;

a determiner responsive to said destination detector for determining whether or not a connection is possible through a computer network;
a route initiator responsive to said determiner for initiating a route for said call through one of said computer network and a switched telephone network.

11. A router as claimed in claim 11, including a monitor for monitoring a quality of service for said call where said call is routed through said computer network and for, where said quality of service falls below a threshold, causing said route initiator to dynamically reroute said call through said switched telephone network.

12. A router as claimed in claim 12, including a signal generator responsive to an output of said monitor for generating an in-band signal when said route initiator reroutes a call.

13. A telecommunications network including a router as claimed in any one of claims 10 to 12.

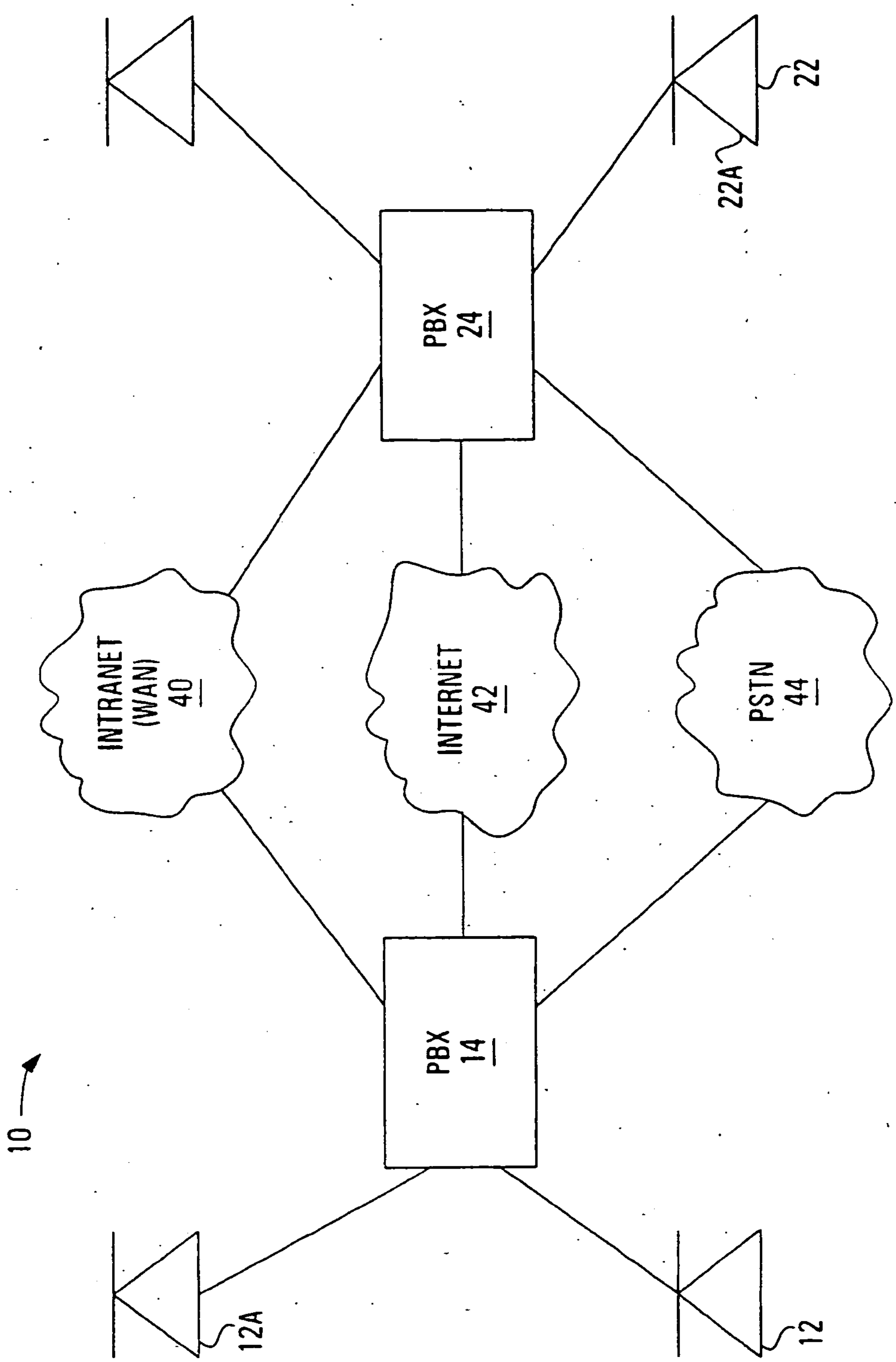
10
INTRANET
(WAN)
40
INTERNET
42
PSTN
44
PBX
24
PBX
14
22
22A
12
12A

FIG. 1

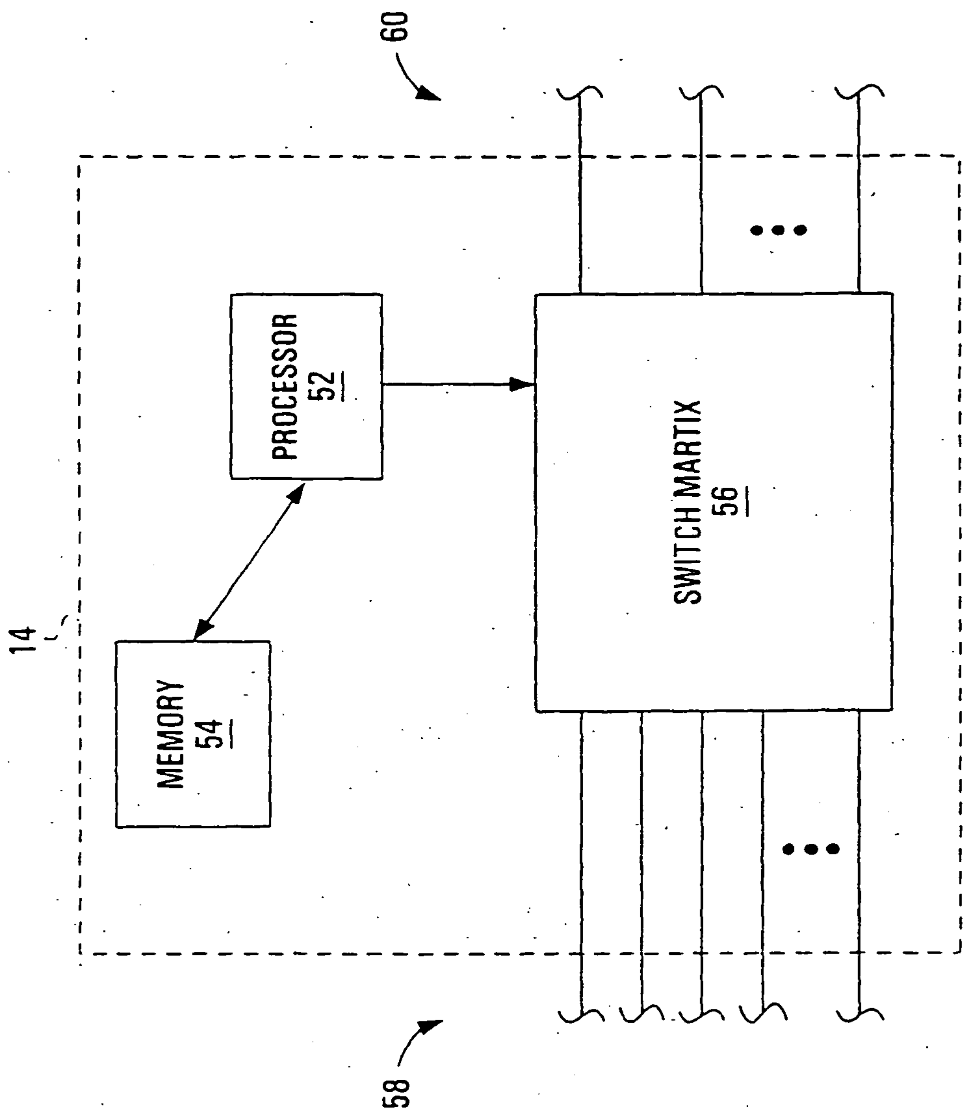

FIG. 2

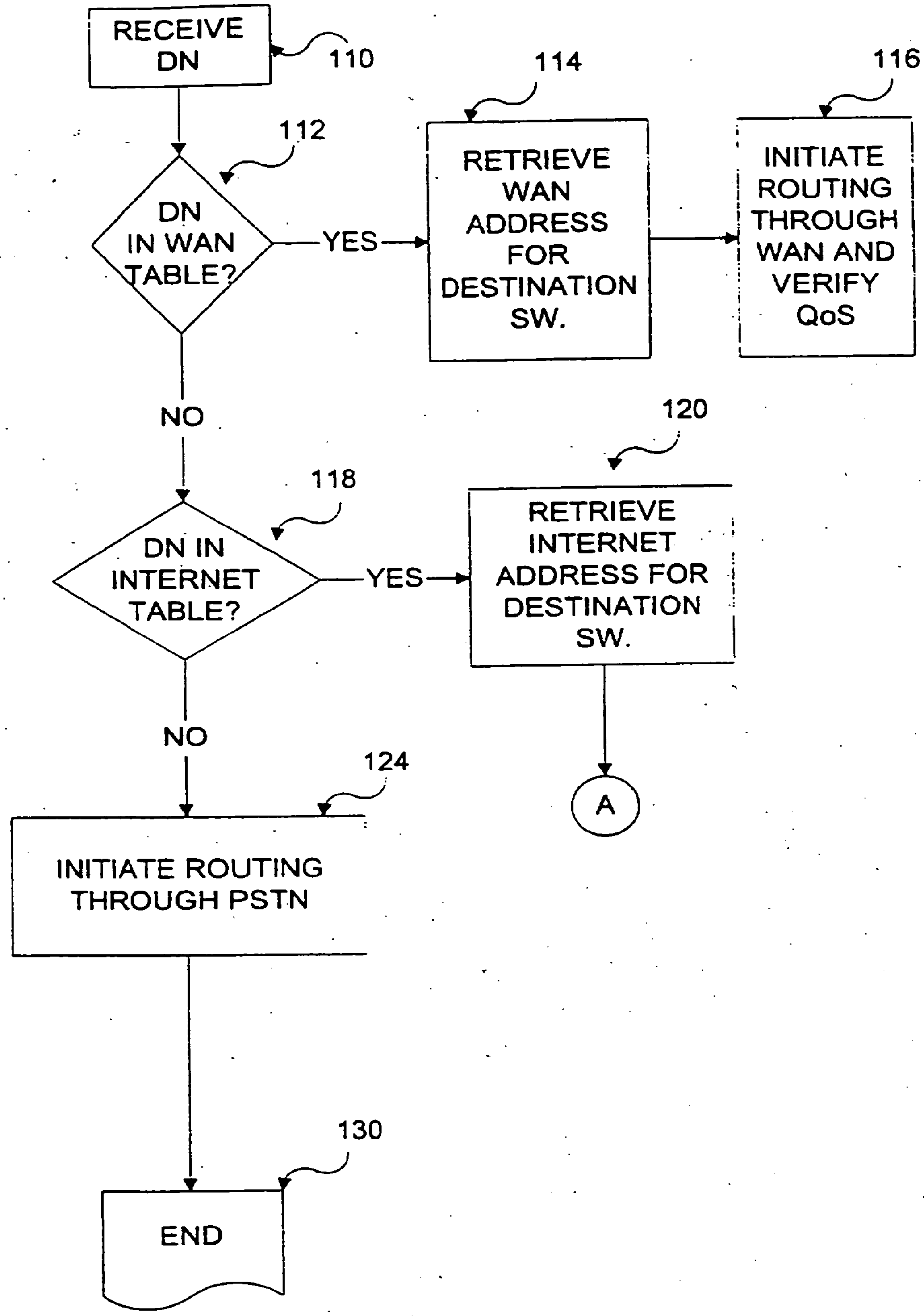
RECEIVE DN
110
DN IN WAN TABLE?
112
YES
RETRIEVE WAN ADDRESS FOR DESTINATION SW.
114
INITIATE ROUTING THROUGH WAN AND VERIFY QoS
116
NO
DN IN INTERNET TABLE?
118
YES
RETRIEVE INTERNET ADDRESS FOR DESTINATION SW.
120
A
NO
INITIATE ROUTING THROUGH PSTN
124
END
130

FIGURE 3a

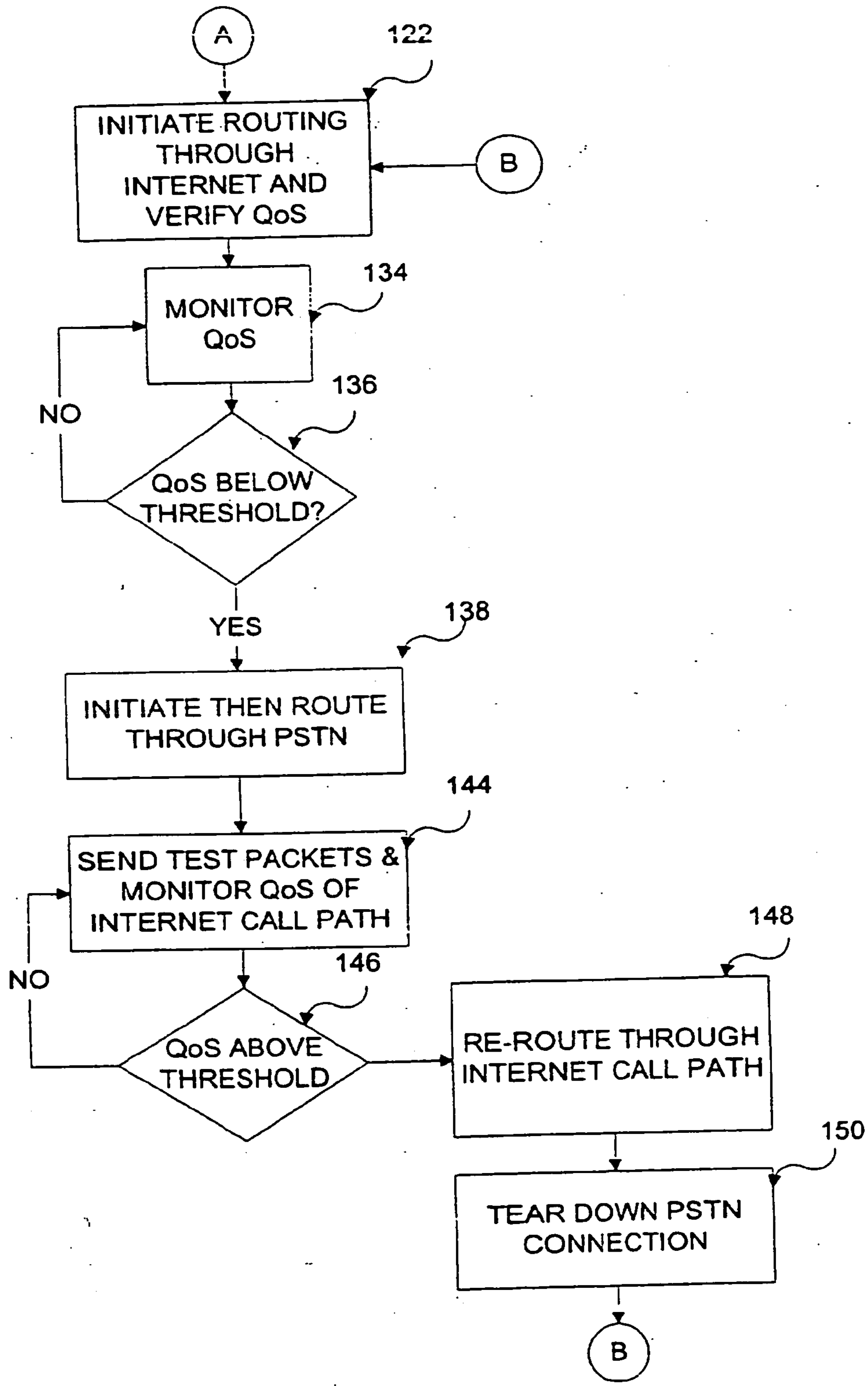
A
122
INITIATE ROUTING THROUGH INTERNET AND VERIFY QoS
B
134
MONITOR QoS
136
NO
QoS BELOW THRESHOLD?
YES
138
INITIATE THEN ROUTE THROUGH PSTN
144
SEND TEST PACKETS & MONITOR QoS OF INTERNET CALL PATH
148
146
NO
QoS ABOVE THRESHOLD
RE-ROUTE THROUGH INTERNET CALL PATH
150
TEAR DOWN PSTN CONNECTION
B

FIGURE 3b

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(71) Applicant: Nortel Networks Limited
Montreal, Quebec H2Y 3Y4 (CA)

(72) Inventor: Cruickshank, Brian
Oakville, Ontario L6L 4N8 (CA)

(74) Representative: Coyle, Philip Aidan et al
F. R. KELLY & CO.
27 Clyde Road
Ballsbridge
Dublin 4 (IE)

(54) Automatic backup trunking for voice over the internet

(57) To reduce telephone toll costs to a user, a PBX preferentially establishes a call to a destination number (DN) over a WAN or the internet. The PBX determines the available connection types available by querying look-up tables for the particular DN. If no alternatives to the PSTN are available, the call is routed over the PSTN. Where a WAN or internet connection is available, the call is then routed over this alternative service. If the Quality of Service (QoS) over the computer network connection falls below a specified threshold, a second parallel connection is made over the PSTN and the call is then transferred to the PSTN. The user is notified of this change in service. During the PSTN connection, the PBX polls the alternative service and, upon the QoS rising above a specified threshold, the call is then routed back to the alternative service and the PSTN connection is torn down. The user is again notified of this change in service.

EP 0 920 176 A3

European Patent Office

EUROPEAN SEARCH REPORT

Application Number
EP 98 30 8869

DOCUMENTS CONSIDERED TO BE RELEVANT			
Category	Citation of document with indication, where appropriate, of relevant passages	Relevant to claim	CLASSIFICATION OF THE APPLICATION (Int.Cl.6)
X	HOUGHTON T F ET AL: "A PACKET TELEPHONY GATEWAY FOR PUBLIC NETWORK OPERATORS" ISS. WORLD TELECOMMUNICATIONS CONGRESS. (INTERNATIONAL SWITCHING SYMPOSIUM),CA,TORONTO, PINNACLE GROUP, 21 September 1997 (1997-09-21), pages 35-44, XP000704453 * page 42, paragraph 6.4 *	1-3,7,8, 10,11	H04M7/00 H04M3/22 H04L29/06
X A	WO 97 27692 A (FIRETALK INC) 31 July 1997 (1997-07-31) * page 12, line 15 - page 17, line 24 *	1,7,9, 10,13 4-6,8,9	
P,X	GRUEN M ET AL: "WEGE AUS DEM DATENSTEU IP-TELEFONIE UND DIE VERKEHRSBEDINGUNGEN IM INTERNET" NET - ZEITSCHRIFT FUER KOMMUNIKATIONSMANAGEMENT,DE,HUTHIG VERLAG, HEILDERBERG, vol. 52, no. 3, 1 March 1998 (1998-03-01), pages 46-49, XP000740475 ISSN: 0947-4765 * page 49, middle column *	1-3,7, 10,11,13	TECHNICAL FIELDS SEARCHED (Int.Cl.6) H04M H04L H04Q

The present search report has been drawn up for all claims

Place of search	Date of completion of the search	Examiner
THE HAGUE	12 July 2001	Megalou, M

CATEGORY OF CITED DOCUMENTS

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12-07-2001

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		EP 0886936 A	30-12-1998

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For more details about this annex : see Official Journal of the European Patent Office, No. 12/82